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CHARLES B. GORDON
THOMAS P. SCHILLER
DAVID B. DEIOMA
JOSEPH J. CORSO
HOWARD G. SHIMOLA
JEFFREY J. SOPKO
JOHN P. MURTAUGH
JAMES M. MOORE
MICHAEL W. GARVEY
RICHARD A. SHARPE
RONALD M. KACHMARIK
PAUL A. SERBINOWSKI
BRIAN G. BEMBENICK
AARON A. FISHMAN

PEARNE & GORDON LLP
ATTORNEYS AT LAW
1801 EAST 9th STREET
SUITE 1200
CLEVELAND, OHIO 44114-3108
TEL: (216) 579-1700 FAX: (216) 579-6073
EMAIL: ip@pearnegordon.com

STEPHEN S. WENTSLER
ROBERT F. BODI
SUZANNE B. GAGNON
UNA L. LAURICIA
STEVEN J. SOLOMON
GREGORY D. FERNENGEL
BRYAN M. GALLO
BRAD C. SPENCER

OF COUNSEL
LOWELL L. HEINKE
THADDEUS A. ZALENSKI

PATENT, TRADEMARK, COPYRIGHT AND RELATED INTELLECTUAL PROPERTY LAW

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March 20, 2006

Mail Stop Certificate of Corrections Branch
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Certificate
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of Correction

Re: U.S. Patent No.: 6,947,570 B2
Issued: September 20, 2006
Title: "TOILET AND METHOD OF OPERATION"
Inventors: Joseph Maisano
Our Docket No.: 33536

Sir/Madam:

A Certificate of Correction under 35 U.S.C. 255 is hereby requested to correct an error in the above-identified patent. Enclosed herewith is a proposed Certificate of Correction (Form No. PTO-1050) for consideration along with the appropriate documentation supporting the request for correction.

It is requested that the Certificate of Correction be completed and mailed at an early date to the undersigned attorney of record. The proposed correction is an obvious one and does not in any way change the sense of the application.

We understand that a check is not required since the error was on the part of the Patent and Trademark Office in printing the patent.

Respectfully submitted,

By Michael W. Garvey, Reg. No. 35878

MWG/jmm
Enclosures

I hereby certify that this correspondence is being deposited with the United States Postal Service as first class mail in an envelope addressed to: Mail Stop Certificateof Corrections Branch, Commissioner for Patents, P.O. Box Alexandria, VA 22313-1450 on the date indicated below.

Michael W. Garvey
Name of Attorney for Applicant(s)

20 Mar 06
Date

Signature of Attorney

MAR 27 2006

PTO/SB/44 (07-03)
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
(Also Form PTO-1050)

UNITED STATES PATENT AND TRADEMARK OFFICE
CERTIFICATE OF CORRECTION

PATENT NO. : 6,947,570 B2
APPLICATION NO. : 09/837,050
ISSUE DATE : September 20, 2005
INVENTOR(S) : Joseph Maisano

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 1, line 26: Please delete "no-called", and insert therefor --so-called--.
Column 1, line 38: Please delete "highs", and insert therefor --high,--.
Column 4, line 53: Please delete "$|S_{1} = S_2|$", and insert therefor --$|S_1|=|S_2|$--.
Column 5, line 37: Please delete "egg.", and insert therefor --e.g--.
Column 5, line 39: Please delete "au", and insert therefor--an--.
Column 5, line 49: Please delete "l", and insert therefor --1--.
Column 6, line 23: Please delete "4*a*" and insert therefor--4 a--.
Column 6, line 30: Please delete "unite", and insert therefor --units--.
Column 6, line 32: Please delete "Past", and insert therefor --Fast--.

MAILING ADDRESS OF SENDER: Michael W. Garvey
Pearne & Gordon LLP
1801 East 9th Street
Suite 1200
Cleveland, Ohio 44114-3108

PATENT NO. 6,947,570 B2

No. of additional copies ⇨ 0

MAR 27 2006

A method for analyzing an acoustical environment and a system to do so

Background of the Invention - Brief Summary of the Invention

The present invention departs from the needs which are encountered in hearing aid technology. Nevertheless, although especially directed to this hearing aid technology, the present invention may be applied to the art of registering acoustical signals more generically.

Current beam formers allow only weighing of incoming acoustical signals according to the spatial direction wherefrom an acoustical signal impinges on an acoustical to electrical converter arrangement.

Besides of generating such spatial angle weighing - beam forming - by means of one respectively ordered acoustical to electrical converter, it is known to provide for such weighing an array of converters, microphones, with at least two microphones. They are located mutually distant by a given distance.

For instance in the hearing aid art it is possible to adapt spatial angle dependent weighing by means of so-called beam forming, so as to eliminate noise from unwanted impinging directions. This enhances the individual's ability to perceive an acoustical signal source situated in a predetermined angular range with respect to the one or - in case of binaural hearing aid - to the two hearing aid apparatuses. Usually by such weighing function acoustical signals are primarily cancelled as impinging from behind the individual.

As current beam formers, especially on hearing aid apparatus, have only an angularly varying response, it occurs in some acoustical environments, as e.g. at a cocktail party, that even if the reception directivity is

high, the speech from a target direction is unintelligible due to superposition of different talkers located in the same direction with respect to the individual carrying the hearing aid apparatus.

It is therefore an object of the present invention to provide for a method for discriminating impinging acoustical signals not only as a function of the angular impinging direction, but also as a function of the distance of an acoustical signal's source from the hearing aid-equipped individual.

More generically, it is an object of the present invention to provide for a method and apparatus for distance-selective monitoring of acoustical signals. It is in a preferred embodiment, as especially for hearing aid apparatus, that the present invention of distance-selective registration of acoustic signals is combined with direction-selective registration of such signals.

By such combining it becomes possible to locate an acoustical source in the acoustical environment, which might be important for non-hearing aid appliances, and for hearing aid appliances it becomes possible to focus reception on a desired source of acoustical signals, as on a specific speaker.

The object of the present invention is realized by a method for analyzing an acoustical environment, which comprises

- registering acoustical signals at at least two reception locations, which are mutually distant by a given distance and generating at least two respective first electric signals representing the acoustical signal;
- calculating electronically from said first electric signals at least one of the distances of sources of

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$$d = \frac{\arg(S_1) - \arg(S_2)}{k} \tag{11}$$

and from (9)

$$r_1 = \frac{\arg(S_1) - \arg(S_2)}{k\left(\frac{|S_1|}{|S_2|} - 1\right)} \tag{12}$$

It can be observed that when the signal comes from the perpendicular of the microphone array axis, some discontinuities occur in the formulas for r_1 because in this case $|S_1|=|S_2|$ and d=0. If the beamforming is a 2[nd] order that eliminates the signal from 90°, there is no need to make a distance calculation in this direction, otherwise a third microphone can be used to perform, in the same way, the distance calculation.

In a preferred form of computation we write:

$$\frac{\rangle|S_1|\langle}{\rangle|S_2|\langle} = \left(1 + \frac{|d|}{r_1}\right) \tag{13}$$

The operator $\rangle\ldots\langle$ thereby represents an average over a predetermined time T during which the signal source may be considered as being stationary with respect to the two microphones 1 and 2. From (13) the distance r_1 becomes

$$r_1 = \frac{|d|\rangle|S_2|\langle}{\rangle|S_1|\langle - \rangle|S_2|\langle} \tag{14}$$

Therefrom, it might be seen that besides of $|d| = p|\cos(\theta)|$ r_1 may again be calculated from the two output signals of the microphones 1, 2. Nevertheless, $|d|$ too may be calculated from these output signals e.g. as will be shown. If we apply to the two signals S_1 and S_2 the function

$$G = \frac{2S_1S_2^*}{|S_1|^2 + |S_2|^2} \qquad (15)$$

there results for kd << 1, i.e. for a distance between the microphones smaller than the wavelength of the respective acoustical signals impinging and further with $d \ll r_1$, i.e. the source being placed in a considerable distance from the two microphones

$$d \approx \frac{\mathrm{Im}[G]}{k}. \qquad (16)$$

Therefrom, there results with (15)

$$r_1 = \frac{|\mathrm{Im}[G]|}{k} \frac{\rangle|S_2|\langle}{\rangle|S_1|\langle - \rangle|S_2|\langle} \qquad (17)$$

It might be seen that r_1 is determined by the two signals S_1 and S_2 at respective frequencies f and with a predetermined distance p and may e.g. be calculated according to (17) too.

In fig. 2 there is schematically shown implementation of the findings which were explained up to now. The two output signals S_1 and S_2 of the at least two microphones 1 and 2 are input to a calculation unit 4, which e.g. according to the formulas (17) and (15) or (12) calculates the distance r_1 and generates accordingly an electric signal $S_3(r_1)$. This signal S_3 is proportional to the distance r_1. The output signal of the calculation unit 4 is applied to the input of an amplitude filter unit 6, which generates an output signal S_4 according to a predetermined filter characteristic or according to a selected or selectable dependency to the magnitude of the input signal S_3 and thus of the distance r_1.

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The output signal S_4 of the amplitude filter unit 6 is applied to an input of a weighing unit 8, as e.g. to a multiplication unit, whereat at least one, e.g. the output signal S_1 of microphone 1 and as applied to a second input of the weighing unit 8, is weighed by the output signal S_4. Thereby, there is generated at the output of the weighing unit 8 a signal S_5 which accords to those parts of signal S_1 which are positively amplified by the amplitude filter characteristics of filter unit 6.

If only the components of S_1 are of predominant interest, which are generated by an acoustic signal source in one predetermined distance, the filter characteristic of amplitude filter 6 is tailored as a band-pass characteristic. Such a band-pass amplitude filter characteristic is e.g. defined by

$$F(f, r_0, r_1) = 1/\left[(r_0 - r_1)^n + 1\right] \tag{18}$$

In Fig. 3 the attenuations F are shown for a predetermined frequency f and for $r_o = 1$, further with $n = 1, 2, 4$ and 8 respectively.

It goes without saying that the amplitude filter unit 6 is most preferably integrated in calculating unit 4 and is only drawn separately in fig. 2 for reasons of explanation.

Considering one of the amplitude filter characteristics of fig. 3 implemented as the filter characteristic of the unit 6 in fig. 2, it becomes clear that only those components of S_1 will be apparent in signal S_5, for which there is valid $r_1 = r_o$, e.g. appropriately scaled for sources with $r_1 = 1$ m.

As additionally shown if fig. 2 it is absolutely possible and often desired to have the filter characteristic of unit

6 made adjustable, so that during operation of the system one can select which area of the acoustical surrounding and with respect to distance shall be monitored.

In fig. 4 there is, still schematically, shown a preferred implementation form of the inventive method and of the inventive system, thereby especially as implied in a hearing aid apparatus or in a binaural hearing aid apparatus set. That signal processing is realized after analogue to digital conversion of S_1 and S_2 and most preferably also after time domain to frequency domain conversion, is quite obvious for the skilled artisan and is also valid at the embodiment of fig. 2. According to the specific needs, the output signal as of S_5 of fig. 2 is respectively reconverted by frequency domain to time domain conversion and subsequent digital to analogue conversion.

According to fig. 4 a matrix of at least two microphones 10 and 12 as of the two microphones of one hearing aid apparatus or of respective microphones at two hearing aid apparatuses of a binaural hearing aid system, which are distant by the respective distance p, generates the respective electric signals S_{10} and S_{12}. The electric output signals S_{10}, S_{12} are amplified, analogue to digital converted and possibly additionally filtered in units 14a and 14b. The output signal S_{14a} and S_{14b} are input to time domain to frequency domain conversion units 16a and 16b, e.g. Fast Fourier Transform units, respectively generating output signals S_{16a} and S_{16b}. In a preferred embodiment and especially for hearing aid appliances the two signals S_{16a} and S_{16b} are fed to a beam former unit 18 where, according to one of the well known calculation techniques, beam forming is realized. As schematically shown in the functional block of unit 18, the output signal S_{18}

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